<u>In the Claims</u>:

Claim 1 (canceled).

Claim 2 (currently amended): A fixed rate speech compression system for processing a frame of a speech signal, the fixed rate speech compression system comprising:

an encoder operable to encode a first part of the frame using common frame based encoding;

the common frame based encoding comprising pitch pre-processing to modify the waveform of the speech signal as a function of classification of the frame;

the encoder operable to select one of a first speech coding mode and a second speech coding mode to encode a second part of the frame, wherein the first speech coding mode uses ~~a two-dimensional vector quantization gain codebook and a two-dimensional code-vector~~ <u>less bits than the second speech coding mode to code a fixed codebook contribution, wherein the first speech coding mode uses a first vector quantizer to jointly code both an adaptive codebook gain and a fixed codebook gain, and wherein the second speech coding mode uses a second vector quantizer to code the adaptive codebook gain and a third vector quantizer to code the fixed codebook gain</u>.

Claim 3 (previously presented): The fixed rate speech compression system of claim 2, where the encoder is operable to continuously time warp the speech signal during pitch pre-processing when the frame is classified as indicative of increased voicing strength.

Claim 4 (previously presented): The fixed rate speech compression system of claim 2, where the encoder is operable to selectively perform continuous time warping of the speech signal during pitch preprocessing to introduce a variable delay of up to about twenty samples.

Claim 5 (previously presented): The fixed rate speech compression system of claim 2, where the encoder is operable to selectively estimate continuous time warping of the speech signal during pitch pre-processing by interpolation with Hamming weighted Sinc interpolation filters.

Claim 6 (previously presented): The fixed rate speech compression system of claim 2, where the encoder is operable to select the first speech coding mode as a function of classification of the frame as at least one of silence/background noise, noise-like unvoiced speech, unvoiced speech, onset speech, plosive speech and non-stationary voiced speech.

Claim 7 (previously presented): The fixed rate speech compression system of claim 2, where the encoder is operable to select the second speech coding mode as a function of classification of the frame as stationary voiced speech.

Claim 8 (previously presented): The fixed rate speech compression system of claim 2, where a frame classified as at least one of background noise and unvoiced speech remains unchanged by pitch pre-processing.

Claim 9 (previously presented): The fixed rate speech compression system of claim 2, where the encoder is operable to time shift the speech signal with pitch pre-processing in a frame classified as predominantly pulse-like unvoiced speech.

Claims 10-33 (canceled).

Claim 34 (currently amended): A method of processing a frame of a speech signal with a fixed rate speech compression system, the method comprising:

encoding a first part of the frame with common frame based encoding, the common frame based encoding comprising:

classifying the frame;

pitch pre-processing to modify the waveform of the speech signal as a function of classification of the frame; and

selecting one of a first speech coding mode and a second speech coding mode to encode a second part of the frame, wherein the first speech coding mode uses ~~a two-dimensional vector quantization gain codebook and a two-dimensional code-vector~~ less bits than the second speech coding mode to code a fixed codebook contribution, wherein the first speech coding mode uses a first vector quantizer to jointly code both an adaptive codebook gain and a fixed codebook gain, and wherein the second speech coding mode uses a second vector quantizer to code the adaptive codebook gain and a third vector quantizer to code the fixed codebook gain.

Claim 35 (previously presented): The method of claim 34, where classifying the frame comprises classifying the frame as a function of pitch correlation information.

Claim 36 (previously presented): The method of claim 34, where pitch pre-processing comprises:

classifying the speech signal as indicative of increased voicing strength; and continuously time warping the frame of the speech signal to introduce a variable delay.

Claim 37 (previously presented): The method of claim 34, where pitch pre-processing comprises:

classifying the speech signal as predominantly pulse-like unvoiced speech; and

time shifting the waveform as a function of an accumulated delay.

Claim 38 (previously presented): The method of claim 34, where pitch pre-processing comprises:

classifying the speech signal as at least one of predominantly background noise and predominantly unvoiced speech; and

resetting an accumulated delay without modification of the waveform.

Claim 39 (previously presented): The method of claim 34, where pitch pre-processing comprises modifying at least one pitch cycle of the speech signal to provide continuous time warping of the speech signal.

Claim 40 (previously presented): The method of claim 34, where selecting the first speech coding mode comprises classifying the frame as at least one of silence/background noise, noise-like unvoiced speech, unvoiced speech, onset speech, plosive speech and non-stationary voice speech.

Claim 41 (previously presented): The method of claim 34, where selecting the second speech coding mode comprises classifying the frame as stationary voiced speech.

Claims 42-51 (canceled).

Claim 52 (currently amended): The fixed rate speech compression system of claim 2, wherein the first speech coding mode uses a two-dimensional vector quantization gain codebook and a two-dimensional code-vector, and wherein fourteen bits are allocated to the two-dimensional vector quantization gain codebook.

Claim 53 (previously presented): The fixed rate speech compression system of claim 2, wherein the second speech coding mode uses two three-dimensional vector quantization gain codebooks.

Claim 54 (previously presented): The fixed rate speech compression system of claim 2, wherein the system is configured to operate at approximately 4 kbits/s.

Claim 55 (previously presented): The fixed rate speech compression system of claim 2, wherein 21 bits are allocated to code linear prediction coefficients.

Claim 56 (currently amended): The fixed rate speech compression system of claim 2, <u>wherein the first speech coding mode uses a two-dimensional vector quantization gain codebook and a two-dimensional code-vector, and</u> wherein the two-dimensional code-vector is selected from the two-dimensional vector quantization gain codebook.

Claim 57 (currently amended): The fixed rate speech compression system of claim 2, <u>wherein the first speech coding mode uses a two-dimensional vector quantization gain codebook and a two-dimensional code-vector, and</u> wherein the two-dimensional vector quantization gain codebook has an adaptive codebook gain and a fixed codebook gain.

Claim 58 (currently amended): The method of claim 34, <u>wherein the first speech coding mode uses a two-dimensional vector quantization gain codebook and a two-dimensional code-vector, and wherein the method</u> further comprising allocating fourteen bits to the two-dimensional vector quantization gain codebook.

Claim 59 (previously presented): The method of claim 34, wherein the second speech coding mode uses two three-dimensional vector quantization gain codebooks.

Claim 60 (previously presented): The method of claim 34, wherein the system operates at approximately 4 kbits/s.

Claim 61 (previously presented): The method of claim 34, further comprising allocating 21 bits to code linear prediction coefficients.

Claim 62 (currently amended): The method of claim 34, <u>wherein the first speech coding mode uses a two-dimensional vector quantization gain codebook and a two-dimensional code-vector, and wherein the method</u> further comprising selecting the two-dimensional code-vector from the two-dimensional vector quantization gain codebook.

Claim 63 (currently amended): The method of claim 34, <u>wherein the first speech coding mode uses a two-dimensional vector quantization gain codebook and a two-dimensional code-vector, and</u> wherein the two-dimensional vector quantization gain codebook has an adaptive codebook gain and a fixed codebook gain.